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REPORT

Filtering of the colour–difference signals in 4:2:2 YUV digital video coding systems

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FILTERING OF THE COLOUR-DIFFERENCE SIGNALS IN 4:2:2 YUV DIGITAL VIDEO CODING SYSTEMS

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Summary

Specifications are proposed for the characteristics of the low-pass filters required at the input and output of the U and V channels of a 4:2:2 YUV digital video coding system. In such a system, the Y (luminance) signal is sampled at 13.5 MHz and the U and V (colour-difference) signals are sampled at 6.75 MHz as specified in CCIR Rec. 601. It has been concluded on both theoretical and practical grounds that these characteristics should be skew-symmetrical about a point of 6 dB attenuation at 3.375 MHz and that they should have a pass-band extending to about 2.6 MHz. It is shown that the use of a skew-symmetric response can reduce the cost and complexity of digital filtering to about one quarter of that required for a non-skew-symmetric response with a similar pass-band and rate of cut-off. Tests are described concerning the subjective effect of the proposed characteristics on displays of very critical test signals.

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FILTERING OF THE COLOUR-DIFFERENCE SIGNALS IN 4:2:2 YUV DIGITAL VIDEO CODING SYSTEMS

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1. Introduction

There is general agreement in the European Broadcasting Union EBU, at least, that the input and output filtering of *U* and *V* signals in a 4:2:2 *YUV* digital video system should be by means of sharp cut-off filters, and that any gradual roll-off in the *U, V* pass-band to prevent ringing near rapid transistions should be inserted in picture monitors or at the input of PAL decoders. (4:2:2 *YUV* implies that the luminance *Y* signal is sampled at 13.5 MHz and that the colour difference *U* and *V* signals are each sampled at 6.75 MHz as specified in CCIR Rec. 601).

The purpose of this Report is to discuss the specification of these sharp-cut filters. For optimum performance, the cut-off frequency of these filters should ideally be at half-sampling frequency i.e. at 3.375 MHz, and the width of the transition band $f_a - f_p$ should be zero, where f_a is the lowest frequency in the stopband and f_p is the highest frequency in the passband as shown in Fig. 1. In practice, however, cost and circuit complexity place a limit on the minimum transition bandwidth that can be employed. Suitable values of f_a and f_p are discussed in Sections 2 and 3.

Other filter characteristics which are considered are the maximum allowable amplitude and group delay variations within the pass-band and the minimum allowable attenuation within the stop-band.

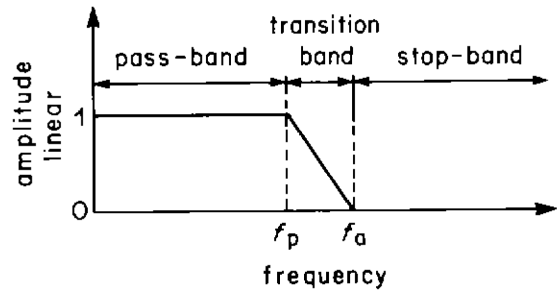

Fig. 1 — Basic features of a low-pass filter.

Both analogue and digital filters have to be considered to cover the different methods, illustrated in Fig. 2, of deriving the digital *U* and *V* signals for a 4:2:2 *YUV* system.

It will be shown that there are very strong arguments for employing an amplitude frequency characteristic which is skew-symmetrical about a point of 6 dB attenuation at 3.375 MHz. Fig. 1 would be of this form if f_a and f_p were symmetrically disposed on either side of 3.375 MHz i.e. if $(f_a + f_p)/2$ were equal to 3.375 MHz.

2. Reasons for using a skew-symmetrical amplitude response

2.1. Maximisation of distortion-free bandwidth

A very important requirement of a *YUV* digital coding system is that the 'distortion-free' bandwidth of the *U* and *V* signals should be kept

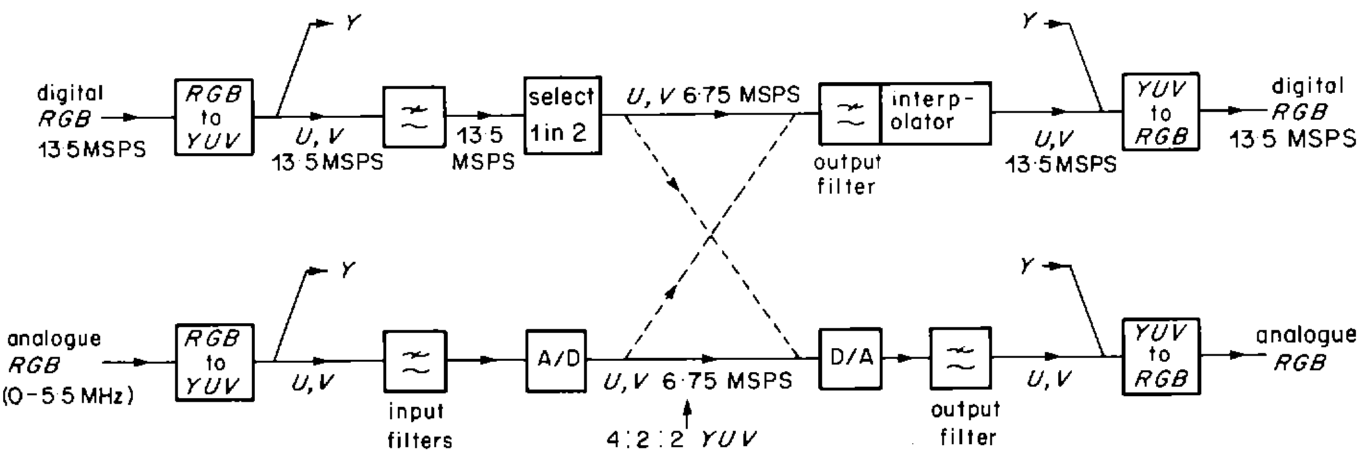

Fig. 2 — Analogue and digital U, V interfacing to a 4:2:2 YUV system. MSPS denotes megasamples per second.

as large as possible. Colour separation overlay (CSO)* operations are particularly critical in this respect. As indicated in Fig. 3, the 'distortion-free' bandwidth is here defined as the bandwidth in which no significant loss of amplitude and also no significant alias components occur i.e. it is the lesser of the two bandwidths labelled loss-free (0 to f_p) and alias-free (0 to $(6.75 - f_a)$). This figure shows the spectra of signals sampled at 6.75 MHz after skew-symmetrical and non-skew-symmetrical input filtering. In this Report, the term skew-symmetrical used without further qualification refers to skew-symmetry with respect to 6 dB attenuation at 3.375 MHz. Both filter characteristics have the same ratio of transition to pass-bandwidth thus normally implying identical circuit complexity. (Skew-symmetic digital filters are much less complex in practice — see Section 2.2). Inspection of Fig. 3 shows that the distortion-free bandwidth is a maximum when f_a and f_p are symmetrically disposed on either side of 3.375 MHz i.e. when $(f_a + f_p)/2 = 3.375$ MHz.

Furthermore, filter theory indicates that the shape of transition band between f_p and f_a for minimum circuit complexity should be approximately skew-symmetrical about the centre of the transition band[1]. This assumes that the same limit has been placed on the peak-to-peak variations within the pass- and stop-bands of the linear amplitude versus frequency characteristic. (If this limit were at 1%, the corresponding limits in dBs would be 0.1 dB in the pass-band and at 40 dB in the stop-band).

Thus, on theoretical grounds, there are good reasons for pre-filtering the *U* and *V* signals with an amplitude characteristic which is at least approximately skew-symmetrical about 3.375 MHz. The same arguments apply to the output filter response.

2.2. Minimisation of circuit complexity with digital filters*

If the input and output filtering of *U, V* signals sampled at 6.75 MHz is in the digital domain, then it is very likely that this filtering will form part of the process of changing between sampling rates of 13.5 MHz and 6.75 MHz. The form of low-pass filter response which is required for these processes is repetitive at intervals of 13.5 MHz and has a cut-off frequency close to

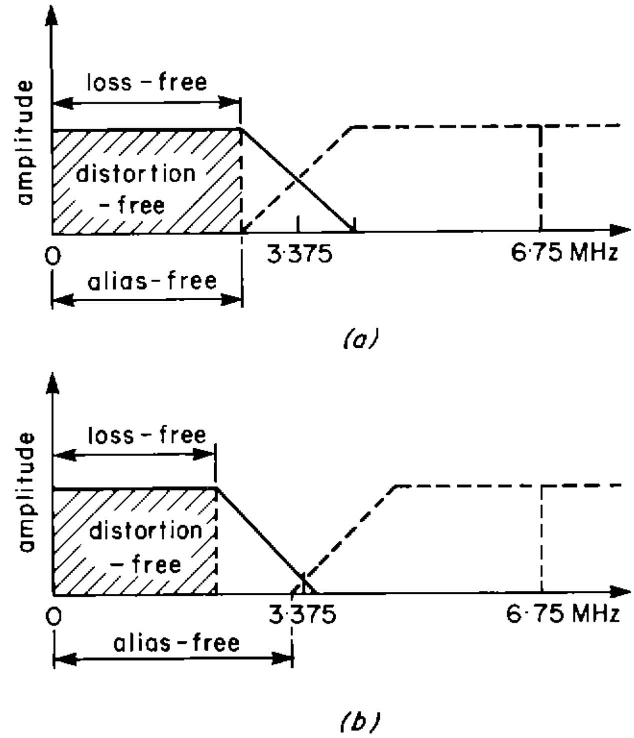

Fig. 3 — Spectra of signals sampled at 6.75 MHz, assuming constant energy per unit bandwidth in input signal.

(a) Skew-symmetric filtering. (b) Non-skew-symmetric filtering.

Wanted components. Alias components.

3.375 MHz as indicated in Fig. 4. (See Appendix, Section 10.1). The corresponding impulse response is of the form shown in Fig. 5 and an instrumentally convenient circuit arrangement is shown in

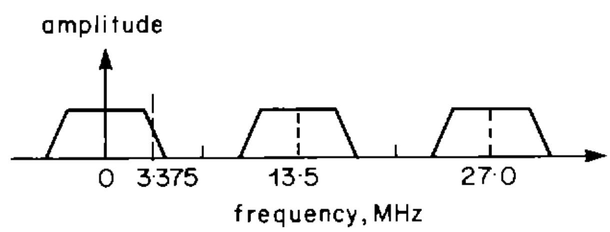

Fig. 4 — General form of frequency response of digital U, V filters.

Fig. 6. (The multiplying factors a_0, a_1, a_2 etc. in these figures will be referred to as coefficients).

Following an input filter, the *U* or *V* signals sampled at 6.75 MHz are obtained by selecting alternate 13.5 MHz samples from the filter as shown in Fig. 7(a). The same filter is suitable as an output filter using the arrangement given in Fig. 7(b). As shown in this figure, the data

* Also known as chroma-key.

* Some aspects of this work are the subject of a patent application.

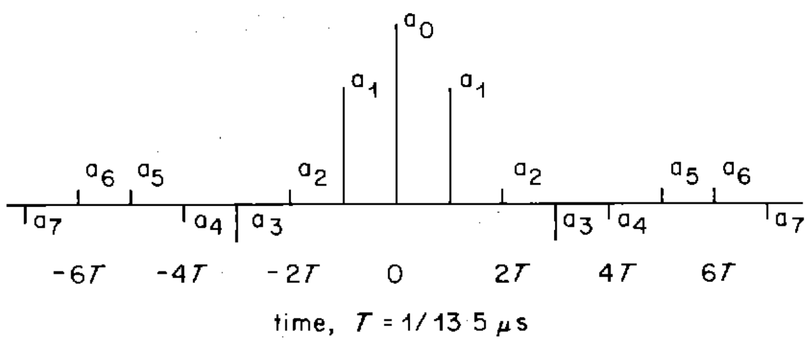

Fig. 5 — General form of impulse response of digital U, V filters.

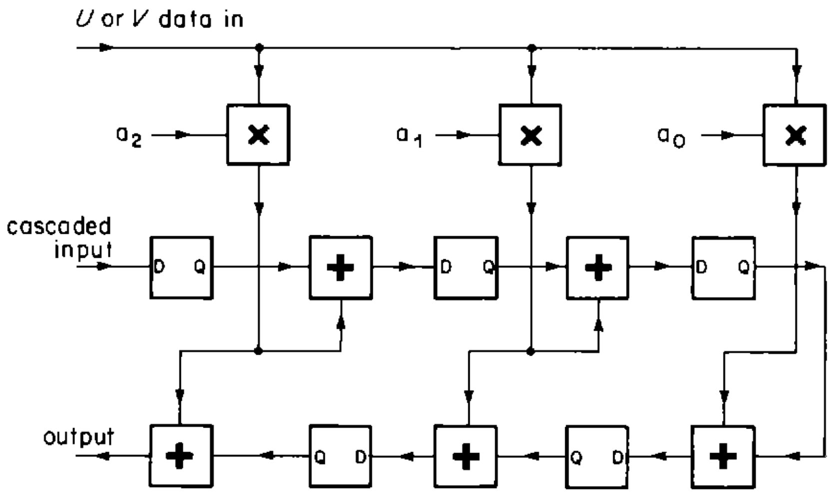

Fig. 6 — Convenient circuit arrangement for digital filters.

Notes: Multipliers typically consist of PROMs + D-type flip flops. All D-type flip-flops are clocked at the input data sample rate.

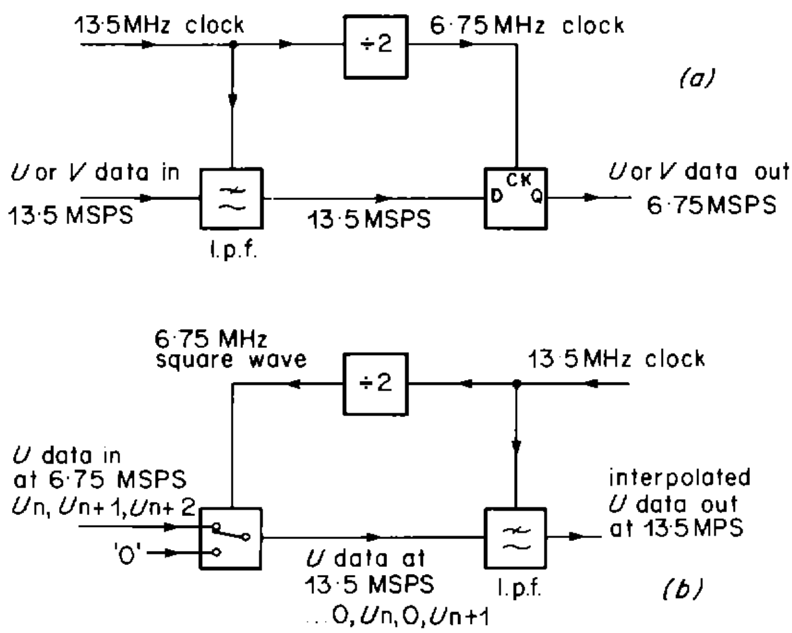

Fig. 7 — Combined filtering and sample rate changing.

MSPS = Mega-samples per second.

(a) Input filter plus 13.5 MHz to 6.75 MHz sample rate changer.

(b) Output filter plus 6.75 MHz to 13.5 MHz sample rate changer.

rate of the 6.75 MHz *U* or *V* signal is increased to the 13.5 MHz rate by inserting samples of value zero between each *U* or *V* sample prior to filtering.

The following discussion shows that with digital filtering, considerable savings in circuit complexity, and hence cost, may be achieved by employing filters whose responses are precisely skew-symmetrical about 3.375 MHz, the importance of this frequency being that it is one quarter of the input data rate. These savings are obtained because the corresponding impulse response has a centre coefficient a_0 which is equal to exactly one half, for an overall filter gain of unity, and the remaining even coefficients a_2, a_4 etc. are all equal to zero.

The most obvious circuit simplifications which can be made with these coefficients are that the multipliers and adders but not the delays, corresponding to a_2, a_4 etc. (see Fig. 6) can be omitted; in addition, the multiplier required for a_o is purely a digital shifting operation. The effect of these simplifications is to approximately halve the cost of the filter for a given duration of impulse response which, in terms of frequency response, implies a given width of transition band.

Less obviously, a further halving of costs can be achieved when a_2, a_4 etc. equal zero because it is then possible, with a modified filter arrangement, to use the same filter for both the U and V signals.

The modified input filter arrangement is shown in Fig. 8. In this figure, the full lines indicate the circuitry required for filtering the U (or V) signal by itself, while the broken lines indicate the additional hardware required for the combined U and V filtering. There are two main alterations to the previously described filter design. Firstly, the processing corresponding to the centre coefficient a_o is performed in a separate path so that all the even coefficients including a_o in the impulse response of the 'LPF' block shown in Fig. 8 are equal to zero. Secondly, the U and V input samples are switched at a rate of 6.75 MHz in such a way that for two simultaneous U, V input data streams

$U_n, U_{n+1}, U_{n+2}, U_{n+3}$ etc.

and

$V_n, V_{n+1}, V_{n+2}, V_{n+3}$ etc.

the data streams sent to the by-pass path and the 'LPF' block have successive sample values given by:

$U_n, V_n, U_{n+2}, V_{n+2}, U_{n+4}$ etc
(via by-pass $-$ a_o coeff.)

and

$V_{n-1}, U_{n+1}, V_{n+1}, U_{n+3}, V_{n+3}$ etc
(via LPF $-$ odd coeffs.)

The resulting 13.5 MHz data from the filter contains multiplexed 6.75 MHz U and V data with successive sample values of the form:

Sample k ..... $a_3 U_{n-3} + a_1 U_{n-1} + a_o U_n + a_1 U_{n+1} + a_3 U_{n+3}$ ....

Sample (k + 1) ... $a_3 V_{n-3} + a_1 V_{n-1} + a_o V_n + a_1 V_{n+1} + a_3 V_{n+3}$ ....

Sample(k + 2).. $a_3 U_{n-1} + a_1 U_{n+1} + a_o U_{n+2} + a_1 U_{n+3} + a_3 U_{n+5}$ ...

Sample (k + 3) .. $a_3 V_{n-1} + a_1 V_{n+1} + a_o V_{n+2} + a_1 V_{n+3} + a_3 V_{n+5}$ ....

These sample values are exactly the same as those that would be obtained by employing separate filters for U and V and then multiplexing alternate samples obtained from these filters.

In a similar way, a single filter can be used at the output of a 4:2:2 YUV system for filtering the multiplexed 6.75 MHz U and V data streams to produce two separate U and V signals each sampled at 13.5 MHz. A suitable arrangement is shown in Fig. 9. Successive sample values of the 13.5 MHz U output data derived from 6.75 MHz sample values U_p, U_{p+1}, U_{p+2} etc. are given by:

Sample k $a_o U_p$,

Sample (k + 1) ... $a_3 U_{p-1} + a_1 U_p + a_1 U_{p+1} + a_3 U_{p+2}$ ....,

Sample (k + 2) $a_o U_{p+1}$,

Sample (k + 3) ... $a_3 U_p + a_1 U_{p+1} + a_1 U_{p+2} + a_3 U_{p+3}$ ....,

It should be noted that 6.75 MHz U and V data streams which are identical to those at the input of the filter can be recovered from the output 13.5 MHz data streams by selecting the appropriate alternate samples.

The combined theoretical and practical advantages of a skew-symmetric characteristic discussed above indicate that a specification for the U and V filtering should be based on such a characteristic. The following sections discuss the optimum parameters of a skew-symmetric response and show that there are no disadvantages compared to non-skew-symmetric responses.

3. Optimum compromise between cost and performance of skew-symmetric filters

3.1. Digital filtering

In the following discussion, the complexity of digital skew-symmetric filters will be defined in terms of the number of multipliers required for the

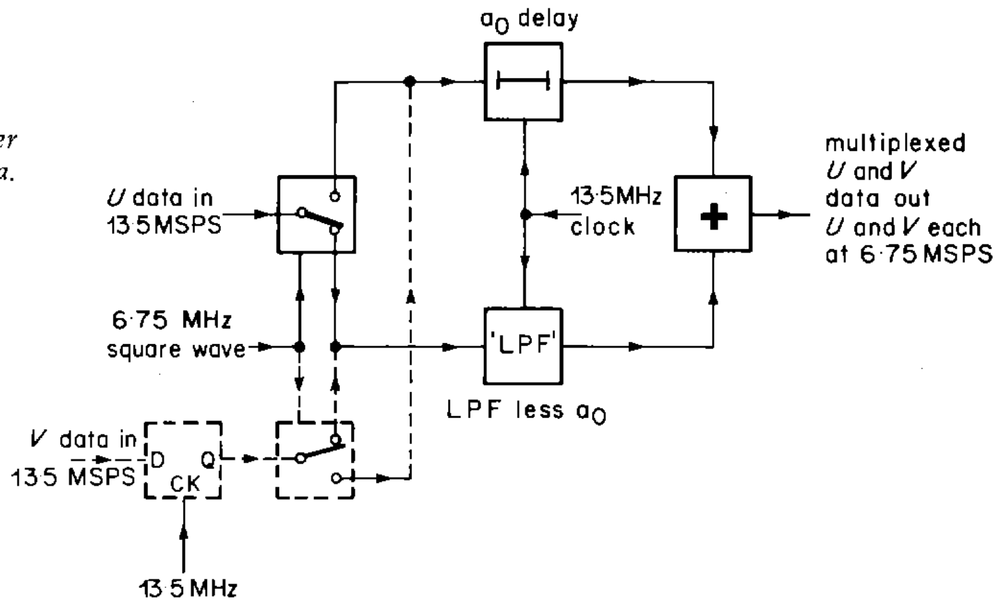

Fig. 8 – Modified input filter for combined U and V data.

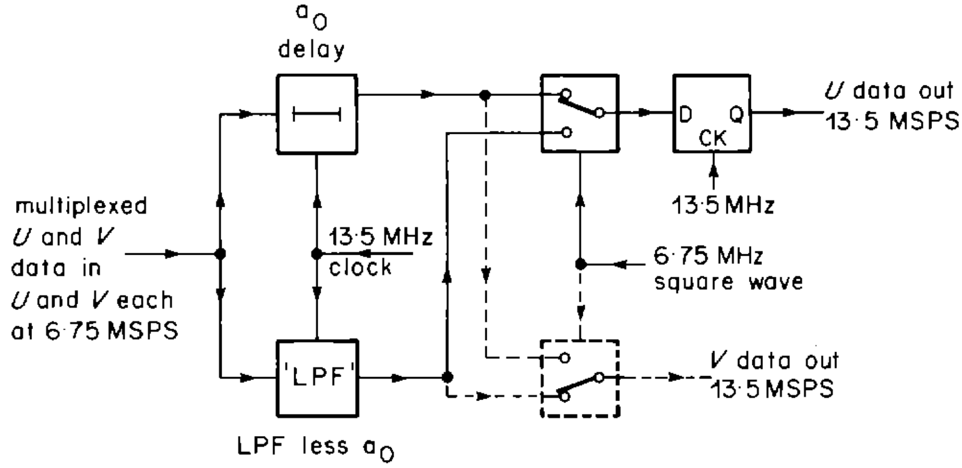

Fig. 9 – Modified output filter for combined U and V data.

odd coefficients. Thus a filter with finite values of a_1, a_3, and a_5 and with all the remaining coefficients except a_0 equal to zero will be referred to as a 3-multiplier filter.

In order to compare the bandwidths obtainable for different numbers of multipliers, it is first necessary to put limits on the allowable amplitude variations in both the pass- and stop-bands. Previous work on the relative visibility of chrominance and luminance errors[2,3] indicates that the allowable limits for amplitude errors in U and V filter responses should be the same as, or slightly greater than, the corresponding limits already proposed for luminance filters[4,5]. On this basis, limits in the pass-band increasing linearly from zero at 0 MHz to ± 0.05 dB at the top of the pass-band f_p should be adequate for the U and V filters in a 4:2:2 YUV system. The corresponding minimum attenuation in the stop-band for precisely skew-symmetric filter is 46 dB which should also be adequate.

Using the above amplitude limits, a family of amplitude versus frequency characteristics have been calculated for different numbers of multipliers per digital filter (See Appendix, Section 10.2). and the results are shown in Fig. 10.

In assessing the cost/performance compromise for the responses shown in Fig. 10, useful information is that the current component cost per multiplier would be about £12 and the number of integrated circuits per multiplier is about 18 using circuit techniques similar to those now in use in the BBC Research Department. Fig. 10 shows that the fractional increase in the width of the pass-band for each additional multiplier decreases rapidly as the number of multipliers increases and a reasonable cost/performance

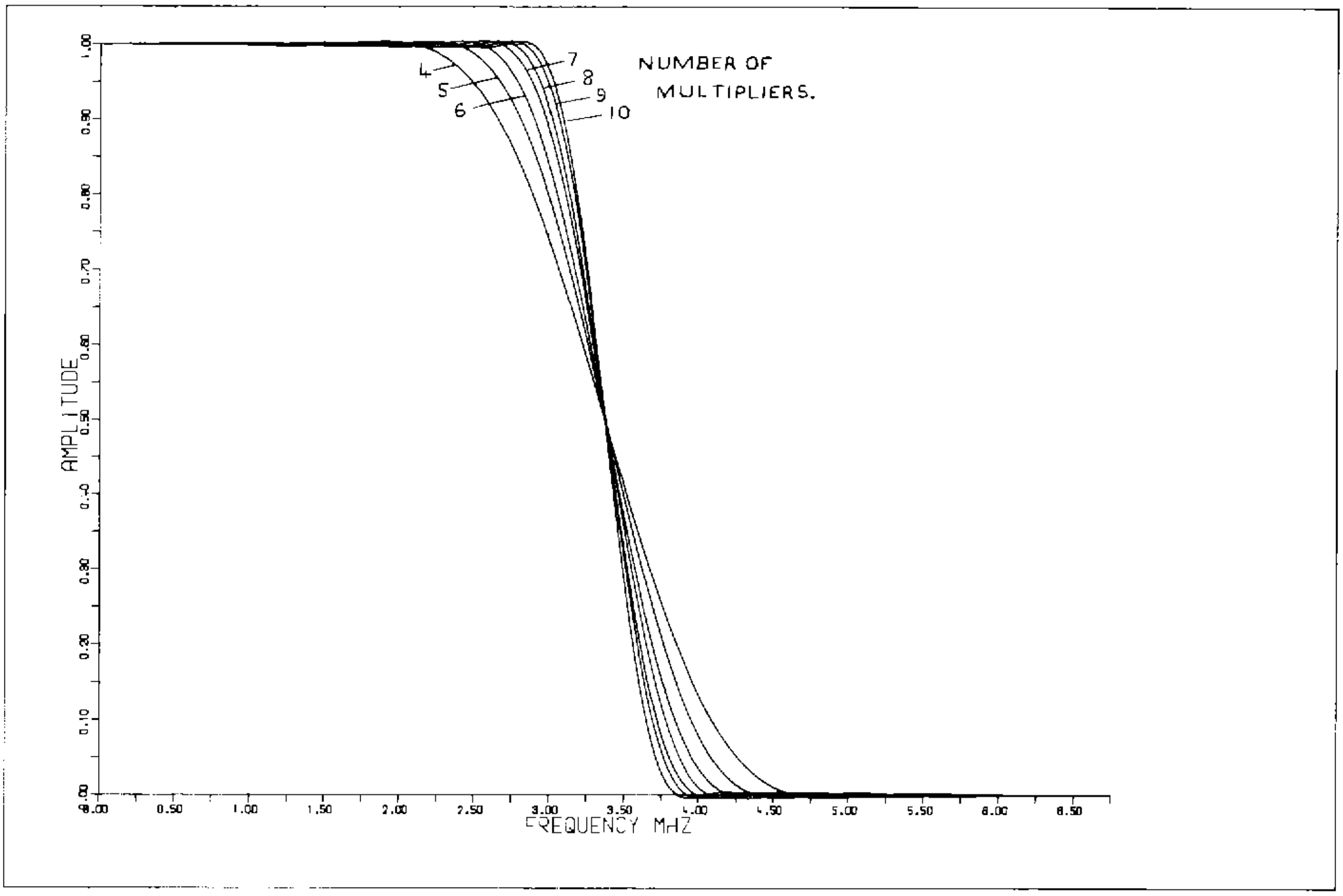

Fig. 10 — Amplitude versus frequency characteristics of U, V LPFs. 4 to 10 multipliers.

compromise would appear to be reached with about 5 to 7 multipliers.

3.2. Analogue filtering

An important factor to be considered in any specification of the *U, V* filter response is that it should be suitable for both analogue and digital filtering (see Fig. 2). At the present time, analogue filters intended for use in the *U* and *V* signal paths of a 4:2:2 *YUV* coding system are available from the principal filter manufacturers in the United Kingdom. The highest performance filters which are available have amplitude responses similar to that shown for 5 multipliers in Fig. 10 but scaled to a slightly lower frequency to give an attenuation of about 12 dB at 3.375 MHz. The complexity of these filters is somewhat less than that required for luminance filters which satisfy the proposed luminance specification[4].

Since the quality of CSO is likely to improve with increase in the *U, V* bandwidth for bandwidths up to and beyond 3 MHz[6], it is desirable to employ as high a bandwidth as possible, a reasonable limit being given by limiting the complexity to that used for luminance filtering. However, a *U, V* filter which is a scaled version of the luminance filter would have twice the delay of the luminance filter and compensation for this difference in delay would be inconvenient. Allowing for the fact that delay is reduced by reducing the rate of cut-off, an analogue *U, V* filter characteristic which is similar to that given for 6 multipliers in Fig. 10 would appear to provide a reasonable compromise between all the factors considered above. (Suitable group delay characteristics are discussed in Section 6).

3.3. The compromise

Taking into account the requirements of both analogue and digital filtering, it appears that the optimum compromise between the cost and performance of skew-symmetric *U, V* filters corresponds to an amplitude versus frequency characteristic similar to that shown for 6 multipliers in Fig. 10. This characteristic is shown separately in Fig. 11. A more precise proposal for the specification of *U, V* filters is given in section 6 following a discussion of subjective tests performed using filters with characteristics as shown in Fig. 11.

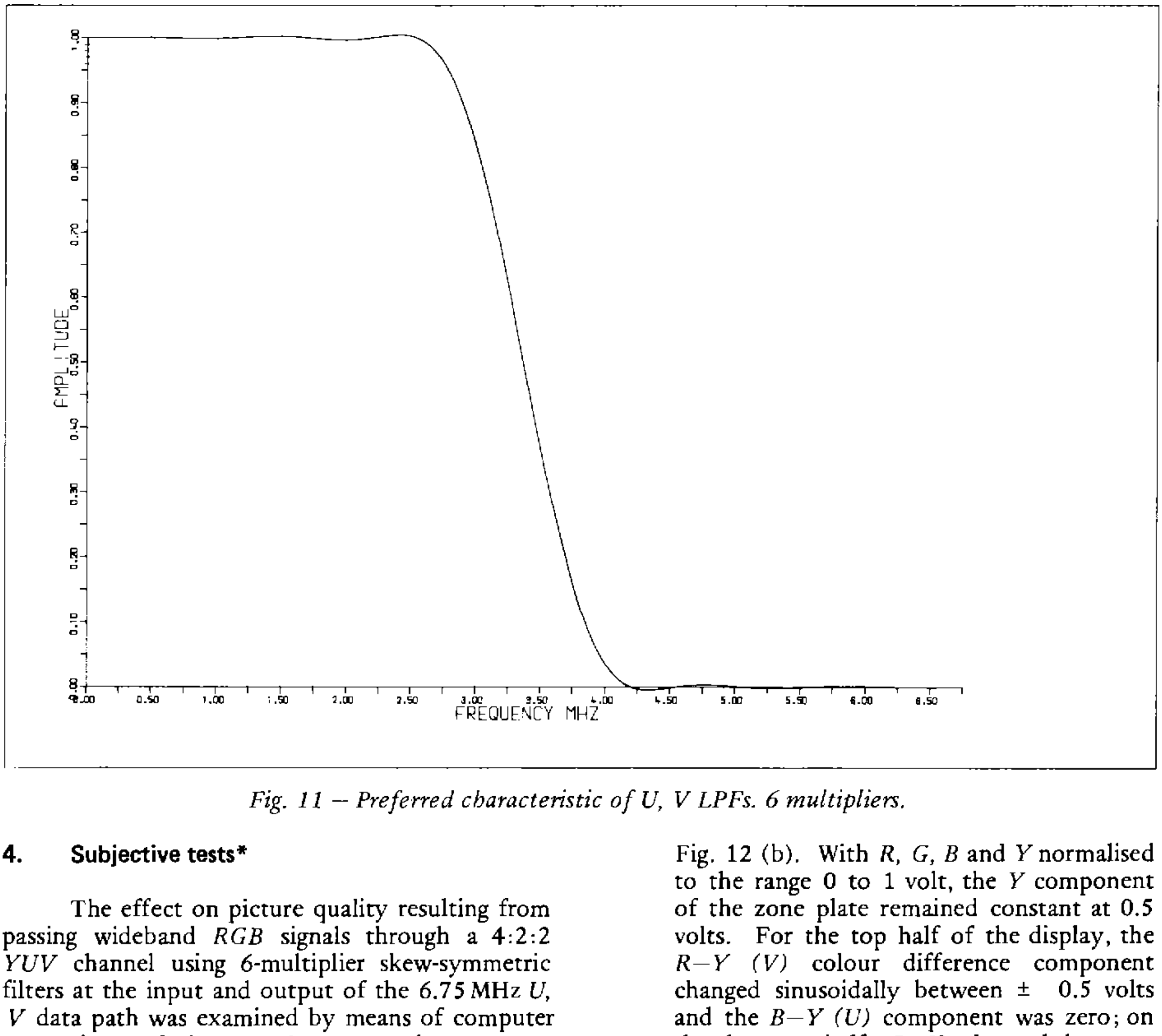

Fig. 11 – Preferred characteristic of U, V LPFs. 6 multipliers.

4. Subjective tests*

The effect on picture quality resulting from passing wideband *RGB* signals through a 4:2:2 *YUV* channel using 6-multiplier skew-symmetric filters at the input and output of the 6.75 MHz *U*, *V* data path was examined by means of computer processing techniques. Two tests pictures were generated, namely:-

(a) A coloured 'chevron' pattern consisting of 128 blocks arranged in 8 rows and 16 columns as shown in Fig. 12(a). All the colours in a 100% saturated colour bar signal including black and white are used in equal numbers. These colours are arranged in a pattern giving all possible combinations of diagonal colour transition.

(b) A circular coloured zone plate pattern covering horizontal frequencies from 0 to 5.5 MHz and vertical frequencies from 0 to 312 cycles per picture height as shown in Fig. 12 (b). With *R*, *G*, *B* and *Y* normalised to the range 0 to 1 volt, the *Y* component of the zone plate remained constant at 0.5 volts. For the top half of the display, the *R–Y (V)* colour difference component changed sinusoidally between ± 0.5 volts and the *B–Y (U)* component was zero; on the bottom half, *B–Y* changed between ± 0.5 volts and *R–Y* was zero.

*The author is grateful to Mr. T.A. Moore for carrying out the computer simulation work described in this Section.

The following three types of processing were examined:

4:2:2 *YUV* processing alone;

4:2:2 *YUV* processing + Gaussian filter;

and Gaussian filter alone,

where 4:2:2 *YUV* processing included the conversions:-

4:4:4 *RGB* ⟶ 4:4:4 *YUV* ⟶ 4:2:2 *YUV* ⟶ 4:4:4 *YUV* ⟶ 4:4:4 *RGB*

(a) (b)

Fig. 12 – Computer generated test pictures. (a) Coloured chevrons (b) Coloured zone plate.

In the above '4' denotes 13.5 MHz sampling; '2' denotes 6.75 MHz sampling.

The Gaussian filtering attenuated the U and V signals by 3 dB at 1.3 MHz (as in PAL coders) and was applied just before the final conversion to 4:4:4 *RGB*.

The resulting pictures were examined in an informal manner in viewing conditions approximately to those in CCIR Rec. 500-2 by about 15 experienced television engineers. The main conclusions derived from these tests were:-

Firstly, with the 4:2:2 *YUV* processing alone, the visibility of the ringing on the coloured chevron pattern caused by the U, V filtering and the visibility of the aliasing on the coloured zone plate caused by the 6.75 MHz sampling were both no worse than the CCIR Rec 500-2 impairment Grade 4 'perceptible but not annoying'. No aliaing effects were visible in the coloured chevron pattern including the diagonal edges and no ringing effects were visible on the coloured zone plate.

Secondly, there was virtually no difference between the pictures obtained with the 4:2:2 *YUV* processing + Gaussian filter compared to the pictures obtained with the Gaussian filter alone; this applied even when the pictures were examined from a distance of less than one picture height. In other words, the ringing and aliasing effects introduced by the 4:2:2 *YUV* processing were eliminated by Gaussian filtering. The colour resolution obtained with either of the above forms of processing was very noticeably better than that obtained after PAL encoding and decoding.

5. Ringing and aliasing obtained with sharp-cut skew-symmetric filters

The main possible objections concerning the use of sharp-cut skew-symmetric filtering of the U and V signals would be that only 6 dB attenuation at 3.375 MHz does not sufficiently attenuate alias components at frequencies close to 3.375 MHz caused by input components close to 3.375 MHz and ringing near chrominance transitions could be annoying. The following arguments indicate that such objections would be unwarranted.

The subjective tests showed that aliasing and ringing effects will not be visible to the home viewer assuming no special effects have been used. (These are considered in the next paragraph). Thus aliasing and ringing will normally be visible only to studio engineers viewing very critical test patterns on monitors which do not include an anti-ringing filter such as the Gaussian filter used in PAL coders. Furthermore, previous experience with digital PAL signals[5] indicates that, even if anti-ringing filters are not used in studio monitors, the degree of aliasing and ringing obtained in the subjective tests will be undetectable on the vast majority of normal pictures*.

Furthermore, special effects such as CSO and picture expansion need not be affected by ringing and alias components close to 3.375 MHz because both these processes require additional filtering for interpolation purposes in their critical signal paths. These interpolation filters are not in

*This has been confirmed by later work using the 6-multiplier filters which have now been constructed in hardware form in equipment designed by C.K.P. Clarke.

the normal main signal path and can therefore be non-standard and provide any necessary degree of alias and ringing suppression.

However, even with special effects, it may be found that the optimum compromise between loss of resolution and picture impairment caused by aliasing is obtained with interpolating filters which attenuate by about 6 dB at 3.375 MHz. This conclusion is based on tests with CSO equipment, operating on 12:4:4 MHz *YUV* signals,[7], in which sharp-cut filters attenuate the *U, V* input signals by 12 dB at 2 MHz and an interpolating filter attenuates the key signal by 6 dB at 2 MHz. With this equipment, lack of resolution compared to that given by CSO operating with 12:12:12 MHz *YUV* signals was definitely noticeable but no significant aliasing effects have so far been observed.

6. Proposed specification of U and V filter characteristics

A proposed specification for the *U* and *V* filters used at the input and output of 4:2:2 *YUV* coding systems is given in Fig. 13. This is based on the response given in Fig. 11 for the reasons given in sections 2 and 3. Features of this specification which have not already been considered are given below.

Although it would obviously be desirable for the analogue specification to be identical to that for digital filtering, differences in associated processing operations indicate that it is sensible to employ different limits in the stop-band region, the reason being as follows:-

At the output of a 4:2:2 *YUV* system, the highest amplitude alias components will normally occur at frequencies close to integer multiples of the sampling frequency of 6.75 MHz. However, with the analogue arrangement of Fig. 2, high attenuation at multiples of 6.75 MHz is automatically provided by the intrinsic sin x/x frequency characteristic of the digital-to-analogue (D/A) converter. Thus it would be unnecessary restriction to specify increased attenuation at 6.75 MHz for analogue filtering. With the digital system shown in Fig. 2, however, the associated D/A converter would be handling the output 13.5 MSPS data streams and would thus give high attenuation at 13.5 MHz instead of 6.75 MHz. With digital skew-symmetric filtering, the requirement for high attenuation at 6.75 MHz is reflected as very low pass-band ripple near 0 MHz which is also very desirable and thus no additional restriction is imposed by the 6.75 MHz requirement. In fact, for a skew-symmetric filter whose response just satisfies the passband specification, the attenuation any-where in the stopband will automatically be at least 5 dB greater than the specified limits given in Fig. 13. The specified

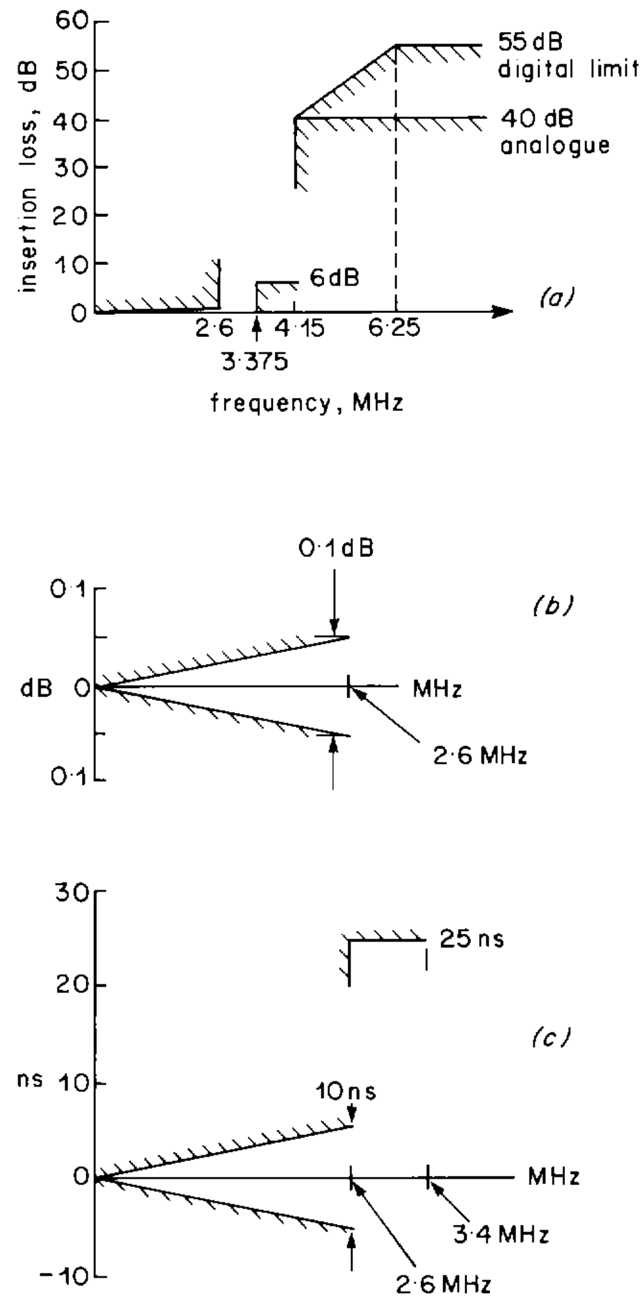

Fig. 13 — Specification of an input or output filter for U, V colour-difference signals sampled at 6.75 MHz.

Note: Insertion loss and group delay are specified relative to their zero-frequency values

(a) Graticule for insertion loss versus frequency response.

(b) Pass-band insertion loss.

(c) Graticule for group delay versus frequency response.

stopband limits are not as high as those achievable with skew-symmetric filters because unnecessarily strict limits could possibly prevent the use of other types of filter whose performance would be perfectly satisfactory in practice.

With regard to the group delay specification shown in Fig. 13, no problem exists with digital filtering as it is convenient to employ filters having a constant group delay at all frequencies. For analogue filtering, limits similar to those employed for luminance filtering have been proposed because it is known that these limits approach the best that can be achieved for reasonable circuit complexity. In addition, however, a limit has been placed on the allowable peak value of the group-delay during the lower-frequency half of the transition-band. This additional limit should ensure adequate symmetry of the impulse response. This requirement is particularly important in CSO processing for which it is very desirable that ringing effects should be minimised while the highest possible resolution is maintained.

The limits given in Fig. 13 for analogue filters are practical limits i.e. they include an allowance for alignment inaccuracies, coil losses etc. The limits for digital filters are theoretical limits which would be achieved in practice for an infinite number of bits per sample. It is worth noting that an error of one half an 8-bit quantum step corresponds to an error of 0.02 dB for a signal occupying the full conversion range. Alternatively, a reduction in the peak-to-peak magnitude of a signal from full conversion range to one half an 8-bit quantum step corresponds to an attenuation of 54 dB.

7. Vertical and temporal filtering of U and V signals

The previous discussions concerning digital filtering apply to the horizontal filtering of U and V signals in processes which halve or double the data rate by omitting or inserting alternative samples along a television scanning line.

Techniques similar to those described in Section 2:2 can be applied to the vertical and temporal filtering required if the overall data rate is to be halved or doubled by omitting or re-inserting alternate line periods or alternate field periods.

For vertical processing, the data selection switches shown in Figs. 8 and 9 would operate at $f_L/2$ instead of $f_s/2$ where f_L is the line-frequency of the television signal and f_s is the sample rate i.e. 13.5 MHz. In addition the delays of one sample period in the low pass filters (see Fig. 6) would be replaced by delays of one line period. The resulting frequency characteristics of the vertical filters would be skew-symmetrical about $f_L/4$ and would be repetitive at intervals of f_L.

8. Conclusions

Specifications have been proposed for input and output filtering of the U and V components of a 4:2:2 *YUV* coding scheme (Y sampled at 13.5 MHz; U and V sampled at 6.75 MHz). These specifications are based on an amplitude characteristic which is skew-symmetric about a point of 6 dB attenuation at 3.375 MHz and has a pass-band extending to 2.6 MHz. The reasons for choosing a skew-symmetric characteristic are that it maximises the distortion-free bandwidth which can be transmitted and that the resulting cost and complexity of digital filtering is about one quarter that required for a non-skew-symmetric characteristic with a similar pass-band and rate of cut-off.

Wide bandwidths in the U and V channels are particularly desirable when key signals for colour separation overlay (CSO) processes are to be obtained from 4:2:2 *YUV* signals.

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10. Appendix

10.1 Function of digital low-pass filters in sample-rate changing between 13.5 MHz and 6.75 MHz

The function of the digital low-pass filters required when the sample-rate of the *U* and *V* signals is changed from 13.5 MHz to 6.75 MHz and vice versa may be explained as follows:

It will assumed that the spectra of the analogue *U, V* or *RGB* signals prior to sampling at 13.5 MHz are flat to a frequency just below 6.75 MHz and close to zero for higher frequencies as shown in Fig. 14(a).

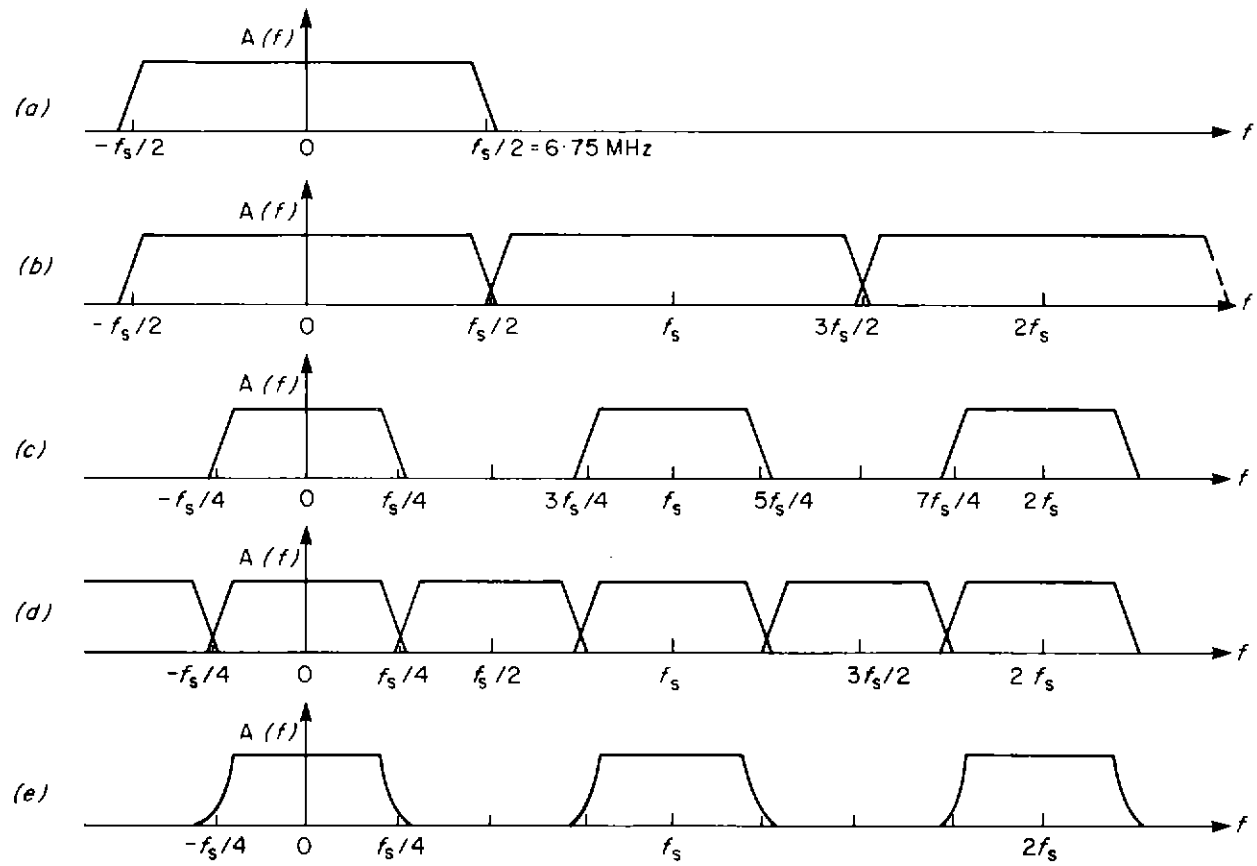

Fig. 14 — Frequency spectra of U, V signals in sample rate changing from 13.5 MHz to 6.75 MHz and vice versa.

(a) Original analogue signals

(b) Input data sampled at 13.5 MHz.

(c) Data sampled at 13.5 MHz after low-pass filtering.
Also frequency characteristic of filters.

(d) Data sampled d at 6.75 MHz after low-pass filtering.

(e) Output data sampled at 13.5 MHz generated from 14(d).

The resulting spectrum of the *U* or *V* data sampled at 13.5 MHz is shown in Fig. 14(b). The input low-pass filters used prior to sub-sampling at 6.75 MHz (see Fig. 7(a)) prevent the overlapping of wanted and alias components after the halving of the sample rate to 6.75 MHz. This filtering should remove components lying in the frequency range between about 3.375 MHz and (13.5−3.375) MHz and corresponding ranges at intervals of 13.5 MHz as indicated in Fig. 14(c) which shows the resulting spectra of the filtered *U* or *V* data streams. The halving of the sample rate to 6.75 MHz by omitting alternate samples of the filtered 13.5 MHz data causes the spectrum of the *U* and *V* data to be repetitive at intervals of 6.75 MHz instead

of at 13.5 MHz as shown in Fig. 14(d).

Considering, now, the process of doubling the sample rate from 6.75MHz to 13.5 MHz, the first step is to insert zero value samples between alternate samples of the U or V data sampled at 6.75 MHz. (see Fig. 7(b)). This process does not affect the spectra of the U and V data streams which remain as shown in Fig. 14(d). The reason why these spectra are unchanged lies in the fact that all the spectra shown in Fig. 14 correspond to instantaneous samples rather than samples lasting for the whole duration of a sample period. With a train of instantaneous samples, the insertion of extra zero value samples obviously has no effect on the spectrum of this waveform. The frequency characteristic of the output filters shown in Fig. 7(b) is similar or identical to that of the input filters and thus the spectrum of the U and V data streams with a sample rate of 13.5 MHz given out by these filters is as shown in Fig. 14(e). This output filtering allows the U and V data streams to be treated as normal data sampled at 13.5 MHz with only low-level unwanted components for frequencies less than 6.75 MHz. As a result they can be combined with other data (e.g. luminance data) sampled at 13.5 MHz with wanted components up to 6.75 MHz.

10.2 Method used for designing the digital filters

The frequency characteristics shown in Fig. 10 are based on an characteristic which has unity gain up to f_p and zero gain above f_a as shown in Fig. 1 but having a cosine-shaped transition band from f_p to f_a. In this transition band:

$$A(f) = \tfrac{1}{2} \cos \left(\frac{\pi (f - f_p)}{f_a - f_p} \right) + \tfrac{1}{2}$$

where $A(f)$ is the amplitude at frequency f.

A smoothly varying transition band such as that provided by a cosine waveform is desirable in order to limit the width of the inpulse response and hence limit the circuit complexity as much as possible. Transition-bands based on the Kaiser-window technique[1] and on the shape given by integrating a cosine-squared pulse waveform were also investigated. Both these other methods gave amounts of passband ripple for a given width of passband which were very similar to that obtained with the cosine-shaped transition-band and will not be further discussed.

The magnitude of the coefficients a_n in the impulse response is given by Fourier analysis of the periodic form of the frequency characteristic as shown in Fig.14(c). For a frequency response of this type with a cosine-shaped transition band which is also skew-symmetrical about $f_s/4$, where f_s is the frequency at which the response is repetitive (i.e. 13.5 MHz for the filters discussed in this Report), the magnitude of coefficient a_n in the corresponding impulse response (see Fig. 5) is given by:

$$a_n = \frac{\mathrm{sinc}(n/2)\mathrm{sinc}(nf_t/f_s)}{2(1 - n^2 f_t^2/f_s^2)}$$

where $\mathrm{sinc}\,(x) = \sin\,(\pi x)/\pi x$

and f_t is the transition bandwidth $= f_a - f_p$.

Note that the values of a_n obtained for all even integer values of n are equal to zero except for $n = 0$ when $a_0 = \tfrac{1}{2}$. It will be found that the sum of the resulting coefficients in the impulse response is equal to unity, indicating unity overall gain for the filter at zero frequency.

It can be seem that this idealised response requires an infinite number of coefficients in the impulse response. To obtain the characteristics shown in Fig. 10 for a finite number of coefficients and hence a finite number of M multipliers in the circuit as shown in Fig. 6, the impulse response was truncated by making $a_n = 0$ for $n > 2M - 1$; the coefficient a_n for which $n = 2M - 1$ was then altered so that the sum of the coefficients was restored to being equal to unity, i.e. $a_{(2M-1)}$ was altered so that:

$$a_0 + 2 \sum_{n=0}^{2M-1} a_n = 1$$

This alteration restored the zero frequency gain to unity.

This technique of adjusting $a_{(2M-1)}$ has the desirable property for video signals that the resulting magnitude of the ripple in the pass-band of the frequency response decreases towards zero as the frequency decreases towards zero.

For a given transition bandwidth, reducing the number of multipliers increases the amount of ripple within the pass- and stop-bands and it also increases the amount by which the coefficient $a_{(2M-1)}$ has to be changed. It was found empirically that the required change in $a_{(2M-1)}$ gave a useful indication of the amount of ripple in the frequency characteristic for a wide range of transition bandwidths. For example, all the characteristics shown in Fig. 10 have a maximum pass-band ripple of 0.05 dB and they all required a change in value of $a_{(2M-1)}$ of between 0.00072 and 0.00078.

For reference, the values of f_t and the corresponding values of f_p used for the characteristics shown in Fig. 10 are given in Table 1.

Table 1 : Pass-bands for different filter complexities and constant pass-band ripple of 0.05 dB

No. of multipliers, M	4	5	6	7	8	9	10
Top of pass-band, f_p MHz	1.9	2.2	2.4	2.55	2.66	2.75	2.82
Transition bandwidth, f_t MHz	2.95	2.35	1.95	1.65	1.43	1.25	1.11

Note that the values of f_p given in Table 1 are the pass-bands corresponding to the precisely unity gain portion of an ideal filter characteristic. When these idealised characteristics are modified by restricting the number of allowable coefficients and the pass-bands are re-defined to allow a certain amount of amplitude variation, the resulting pass-bands are somewhat greater than those given in Table 1. For example, the proposed 6-multiplier filter, based on a value of f_p = 2.4 MHz, has a pass-band of about 2.6 MHz for a pass-band ripple of less than 0.05 dB.

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